REMARKS/ARGUMENTS

This communication is in response to the Office Action of June 15, 2005. Accordingly, this response is accompanied by a request for a three-month extension of time along with the required fees.

Further, this response is accompanied with a Request for Continued Examination with the appropriate funds.

Claim Amendments

In this response, claims 1, 33, 45-49, 52 and 55 have been amended. No claims have been cancelled or added.

Claim 1 has been amended to recite that the method comprises receiving a user adjustable digital loudness normalization control signal from a user during operation for dynamically adjusting compression characteristics for controlling the configuration of the input/output characteristic for loudness normalization. Support for this claim amendment is in lines 14-17 on page 13 of the application as originally filed. This is also discussed in pages 15-19 and shown in Figures 7A-9 of the application as originally filed.

Claim 33 has been amended to recite that the signal processing apparatus comprises a loudness normalization adjustment stage including a signal controlling device that is manipulated by a user during operation to provide a user adjustable digital loudness normalization control signal for dynamic loudness normalization by adjusting compression characteristics of an input/output characteristic of the signal processing apparatus. Support for this claim amendment is in lines 14-17 on page 13, and in claim 1 of the application as originally filed. This feature is also discussed in pages 15-19 and shown in Figures 7A-9 of the application as originally filed.

Claims 45-49 have been amended to properly refer to the antecedent "input/output characteristic" which has been introduced into amended claim 33.

Claims 52 and 55 have been amended to properly refer to the antecedent "input/output transfer function stage".

Response to Arguments

The Applicant thanks the Examiner for his comments on pages 2 to 4 of the Office Action with regards to the Applicant's comments made in response to the previous Office Action. The Applicant would like to make some comments with respect to the Examiner's comments from the second last line of page 3 to the second line of page 4.

Firstly, the Applicant respectfully notes that audio filtering and audio compression are two different separate concepts. Filtering an input audio signal to produce an output audio signal involves the elimination of certain frequency components from the input audio signal; these frequency components will not be present in the output audio signal. Compression on the other hand refers to the relationship between the sound level of the input audio signal and the sound level of the output audio signal, and more specifically how different amounts of gain are applied to the input audio signal depending on the level of the input audio signal to produce the output audio signal. An illustration showing one example of compression is shown in Figure 8 of the subject application. The use of filtering will not result in compression since the same amount of filtering is applied regardless of the input level of the input audio signal. Conversely, the use of compression will not result in filtering since only the amount of gain is being varied, no frequency components are being eliminated.

Secondly, the concepts of input compression and output compression do not refer to looking at a system from an input side or an output side. Rather, input compression and output compression refer to particular signal processing configurations in which compression is controlled based on audio signal information taken at different points in

a signal processing apparatus. Figures 4A illustrates an output compression configuration (the input to the level detector which decides the amount of gain is taken at the output signal processing apparatus) and Figure 4B illustrates an input compression characteristic (the input to the level detector which decides the amount of gain is taken closer to the input of the signal processing apparatus). These concepts are also described on pages 10 and 11 of the application as originally filed.

Claim Rejections – 35 USC § 103

In the Office Action, the Examiner rejected claims 1-40, 42, 43 and 45-57 under 35 USC 103 (a) as being unpatentable over U.S. 5,892,836 by Ishige (hereafter referred to as Ishige) in view of U.S. 6,104,822 by Melanson (hereafter referred to as Melanson).

Regarding claims 1, 13 and 33, the Examiner argued that Ishige discloses a method of generating an analog acoustic output signal from an acoustic input signal in accordance with a configurable input/output characteristic, the method comprising the steps of: (a) converting the acoustic input signal into a digital acoustic input signal; (b) transforming the digital acoustic input signal into one or more frequency domain input signals, and (c) detecting the magnitude of each of the one or more frequency domain input signals.

However, the Examiner noted that Ishige does not explicitly disclose the steps of: (d) receiving a user adjustable digital loudness normalization control signal for dynamically controlling the configuration of said input/output characteristic; (e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal; (f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value; (g) transforming the one or more frequency domain output signals into a digital acoustic output signal; and (h) converting the digital acoustic output signal into the analog acoustic output signal.

The Examiner also argued that Melanson discloses a digital signal processor hearing aid with a program selector switch that is preferably manipulable by a user to allow the user to dynamically select one of several digital signal processing means to invoke in a particular listening environment. In dealing with these environments, each of the processing means may implement such functions as compression, noise compensation, feedback cancellation etc.

The Examiner is of the opinion that applying this environmental selection switch to Ishige would allow the user to conveniently alter the characteristics of Ishige's hearing aid to further assist the user in various environments. The Examiner further argued that it would be obvious to one of the ordinary skill in the art at the time the invention was made to apply the adjustable features of Melanson to the hearing aid of Ishige.

In response, the Applicant notes that if one were to combine the Ishige and Melanson references, the combination cannot change the principle of operation of one of these references (see sect. 2143.03 of the MPEP). The Applicant submits that this would clearly happen if these references were combined.

Ishige teaches a digital hearing aid comprising a hearing compensating circuit having a transposed transversal filter, an analyzer for frequency-analyzing an input signal, a memory storing hearing characteristics of a person to be fitted with the hearing aid, and a controller receiving a frequency analysis result of the input signal and the hearing characteristics for deriving coefficients for the transposed transversal filter and supplying the derived coefficients to the transposed transversal filter. In particular, Ishige teaches the use of a plurality of linear phase filters which are weighted, based on the frequency analysis results and the hearing characteristics of the user, to come up with the transposed transversal filter.

Ishige does not teach a user input means that the user can use during operation to control the performance of the hearing aid during operation. The only information

obtained from the user is the user's hearing characteristics, which are stored in the memory of the digital hearing aid when the hearing aid is manufactured. This is clearly a static operation, and does not allow the user to later provide a control signal to the hearing aid to dynamically control the performance of the hearing aid during operation. Ishige simply teaches a digital hearing aid having variable hearing compensating characteristics that depend on the level of the input audio signal.

In general, Melanson teaches a program selector switch that the user can set to one of a limited number of digital signal processing means that implement a particular processing strategy designed for a particular situation. Melanson further teaches that one of the digital processing means may be designed to compensate for noisy environments while another may be designed for quiet environments and that in dealing with each of these environments, each of the processing means may implement such functions as compression, noise reduction, feedback cancellation, etc. (see column 8, lines 30-49). A program selector switch allows the user to switch from one processing strategy to another (see column 19, lines 22-38).

Accordingly, if one were to combine these references, the principle of operation of the Ishige reference would definitely change and would not function as originally intended. Ishige teaches deriving coefficients for the transposed transversal filter based on a frequency analysis result of the input signal and the hearing characteristics of the user (obtained by a static measurement and programmed during manufacturing), and using these coefficients to process the input audio signal. Ishige only envisions one processing strategy to process the input audio signal and does not even consider the possibility of altering the processing strategy based on user input. Combining these features from the Melanson patent would definitely change the operation of the Ishige reference from what was originally intended.

Furthermore, the Applicant submits that even if one were to combine the Ishige and Melanson references, the combination does not teach or suggest all claim limitations of the subject application (see sect. 2143.03 of the MPEP).

Upon closer inspection of the Melanson reference, what Melanson really teaches is varying the type of filtering that is performed by each of these processing strategies. For instance, in lines 11-15 on column 9, Melanson states:

> *"In actuality, what is often changed in going from one processing means to another is the number of bandpass signals into which the input digital signal is divided, and the bandwidths of these bandpass signals. Thus, by changing strategies, the hearing aid of the present invention is in effect changing the specifications of the filter banks of the digital signal processing means 50. This will become more clear as the invention is described in greater detail."*

In line 67 on column 9 to line 3 on column 10, Melanson states:

> *"the system provides for a plurality of programs that the user can switch between, it is possible to change filter structures by selecting different programs."*

In line 9 on column 9 to line 13 on column 10, Melanson states:

> *" One of the motivations for the various filter bank embodiments is to achieve high frequency resolution, especially at low frequencies, without incurring large delay through the system. This issue will be discussed in conjunction with the various embodiments."*

The majority of the description, i.e. columns 10-16, of Melanson discusses how the filterbank structure is varied based on the user input.

In addition, although Figure 2 of Melanson shows a multi-band compressor 202, upon reading this reference from line 63, column 16 to line 18, column 18, and reviewing the corresponding Figures 22 and 23, it is clear that the compressor block is static. Melanson does not teach varying the parameters of the compressor 202 based on user input. In fact, in column 16, lines 63-64, Melanson states that the structure of one band of the multi-band compressor shown in Figure 22 is repeated for every band. This structure includes an instantaneous power estimate block 2201, a Log block 2202, a Log smoother block 2203, a gain calculation block 2204 and a summer 2205. Figure 23 shows the structure of the Log smoother block 2203 which includes a subtraction block 2301, a lookup function table 2302, summer block 2304 and 2305, a smoothing filter

state block 2306 and a smoothing coefficient generator block 2307. It is clear that none of these blocks use a user input to vary characteristics of the compression.

In contrast, with regards to claim 1 of the subject application, the Applicant notes that this claim recites receiving a user adjustable digital loudness normalization control signal from a user during operation for dynamically adjusting compression characteristics for controlling the configuration of said input/output characteristic for loudness normalization, as well as, for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal. This allows the user to select the shape/curve of the input/output characteristic according to their preference (see Figures 8-9 of the subject application), and adjust this characteristic over time if their hearing characteristics change.

It is clear that this feature is not taught in either the Ishige or the Melanson references when taken alone or in combination. As mentioned earlier, Ishige does not teach using a user input to alter the performance of his hearing aid, and although Melanson teaches using a user input, he does not teach using the user input to alter compression characteristics for altering an input/output characteristic for loudness normalization. Rather, it is very clear that Melanson only teaches altering the structure of the filterbanks, such as the number of bands, and the like based on the signal processing strategy that is selected by the user. Accordingly, if one of the preprogrammed static processing methods in the Melanson device does not match the user's hearing characteristics, which may change over time, the user cannot modify the performance of the Melanson device to address the change in their hearing characteristic. The user is restricted with the Melanson device to simply selecting the processing method that the least unacceptable performance.

The Applicant further submits that the user adjustable digital loudness normalization control (LNC) signal recited in claim 1 is used in a different manner and provides different results than the program selector switch taught by Melanson. The program selection switch of Melanson can only be used to select between a limited number of programs. Furthermore, these programs are optimized for operation in different listening environments and cannot be altered by a user input to take into account changes in the listener's hearing characteristics over time.

As explained in the passage at lines 4-12 on page 19 of the subject application, the claimed invention permits the use of minimal measurements during an initial fitting to first measure the user's auditory characteristics, and then allows the user to adjust the input/output characteristic of the loudness response of their hearing aid according to the user's preferences by adjusting compression characteristics. This allows for the elimination of the time-consuming, and laborious task of measuring loudness data for the user, which is typically an inaccurate measurement. Furthermore, loudness judgment is highly subjective and in some cases loudness ratings can change from day to day (for a variety of reasons). Accordingly, a user most likely will want to adjust the compression characteristics of the hearing aid during use.

The user will be able to do so based on the claimed subject invention in a dynamic manner during operation. For example, the user can use the user adjustable digital loudness normalization control signal to dynamically adjust compression characteristics according to their preference. A gain value is then determined in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal. Accordingly, the user control signal recited in claim 1 allows the user to directly control the compression characteristics of the input/output characteristic as they see fit and they are not limited to select between one of a small number of pre-programmed static programs as taught by Melanson. Accordingly, the user control recited in claim 1 allows the user to provide input to compensate for changes in the user's loudness perception over time. This is not possible with the Melanson device.

With respect to claim 13, the subject application recites receiving N user adjustable digital loudness normalization control signals from a user during operation for dynamically controlling a configurable composite input/output characteristic for loudness normalization, each of the loudness control signals corresponding to one of a number of frequency domain input signals, and determining N gain values, each of the gain values corresponding to one of the frequency domain input signals and each of the gain values being determined in response to one of the frequency domain input signals and to one of the user adjustable digital loudness normalization control signals.

These separate inputs allow the user to separately adjust the input/output characteristics of each channel in a multi-channel amplification device for loudness normalization (see pg. 17, lines 12-18 of the subject application as originally filed). This multi-control allows the user to control loudness normalization, and hence to affect the gain, of each channel independently of the others and therefore allows a much wider range of adjustment.

It is clear that such a multi-channel control embodiment is more flexible than the program selector switch taught by Melanson. Melanson simply teaches the use of one control signal that can be provided by the user. Furthermore, the program selector switch taught by Melanson limits the user to selecting between a small number of programs preset into the device. In addition, although Melanson teaches multi-band processing, the user is not allowed to separately control any parameters used in the different bands.

With regards to claim 33, a similar argument can be made as was made for claim 1.

With regards to claims 2, 4, 5, 7, 14, 18, 19, 21, 22, 25, 31, 32, 34-40, 51, 53, 54, and 57, the Applicant respectfully submits that these claims recite the use of the user adjustable digital loudness normalization control signal for various aspects of altering the compression characteristics of the input/output characteristic. As previously

mentioned, the Applicant submits that the Ishige reference does not teach a user input. Further, if the Melanson reference could be combined with the Ishige reference, the combination would not teach using a user input signal in the manner recited in these claims. As explained previously, Melanson's use of compression is fixed and is the same for each frequency band. Melanson does not the user input to vary his compression algorithm.

Accordingly, the Applicant respectfully submits that claims 1, 13 and 33 of the subject application are novel and inventive over the cited references and should be allowed. Furthermore, since claims 2-12, 14-32, and 34-40, 42, 43, and 45-57 depend either directly or indirectly on one of claims 1, 13 or 33, and introduce other patentable features, the Applicant respectfully submits that these claims should also be allowed.

CONCLUSION

In view of the foregoing comments, it is respectfully submitted that the application is now in condition for allowance. If the Examiner has any further concerns regarding the language of the claims or the applicability of the cited references, the Examiner is respectfully requested to contact the undersigned at 416-957-1603.

Applicant respectfully requests that a timely Notice of Allowance be issued in this case.

Respectfully submitted,
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